DATA COMMUNICATIONS

FACILITIES, NETWORKS, AND SYSTEMS DESIGN

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A Wiley-Interscience Publication
JOHN WILEY & SONS
New York • Chichester • Brisbane • Toronto

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34. E. R. Cacciamani and K. S. Kim, "Circumventing the Problem of Propagation on Satellite Data Channels, *Data Communications*, July–August, 1975, McGraw-Hill. New York.
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SEVEN

MULTIPLEXING AND CONCENTRATION TECHNIQUES FOR LINE SHARING

Obtaining a cost-effective teleprocessing network is postulated on efficient utilization of the communication links and processing equipment. A variety of line sharing devices and procedures are commonly used for this purpose. In this chapter, various functional and economic aspects of frequency-division multiplexing (FDM), synchronous time-division multiplexing (STDM), statistical time-division multiplexing (STATDM), message switching concentration (MSC), and line (or circuit) switching techniques are discussed.[1,2] Also considered are recently developed sharing techniques known as packet switching and inverse multiplexing. The motivations for line sharing stem from economies of scale in the cost of bandwidth and from the traffic smoothing effect that such devices produce when serving a large terminal population characterized by unscheduled requests for service.

The discussion of these techniques includes a detailed distinction between *multiplexing* and *concentration,* two terms often (and unfortunately) used synonymously. It is shown that FDM and STDM are examples of conventional multiplexing, whereas message switching, packet switching, and line switching usually illustrate concentration. Statistical time-division multiplexing is shown to be a hybrid line sharing scheme embodying certain features of both concepts. Thus it is often referred to as statistical multiplexing.

The first part of the chapter is devoted to a functional explanation of the above-noted techniques. The rest is concerned with applications and systems design situations involving multiplexing and concentration techniques. The application section focuses on important economic factors pertaining to the selection and use of the various methods. The role of line sharing devices in contemporary common carrier and end-user networks is also considered. The economic and technical aspects of these contrasting application environments are emphasized to illustrate the multiplicity of uses for line sharing devices.

The concluding portion of the chapter introduces system design considerations by illustrating precisely how the decision to use multiplexers or concentrators in a typical computer–communication network is implemented. Various techniques for geographically positioning multiplexers and concentrators to minimize total costs are presented. The use of one of these procedures is demonstrated, using a typical design problem as a case study.

Although the theoretical basis for many of these approaches may be found in well-understood concepts of conventional voice telephony, the

[1]Line switching concentration is sometimes also referred to as space-division multiplexing.
[2]Some of the material in this chapter was originally discussed by Doll in Reference [31].

myriad regulatory and economic nuances of today's unsettled communications environment and the unique requirements of the computer industry account for the recent surge of interest in line sharing techniques by noncarrier users. Before the famous Carterfone decision of 1968, such concepts were of concern mainly to the common carriers. Then came permission for complete interconnection and a subsequent realization by end users that costs could be appreciably reduced by employing relatively simple multiplexing devices.

Of the line sharing methods already noted, FDM and STDM are by far the most prevalent in contemporary end-user networks. However, falling minicomputer prices, cost-conscious data users, and a continuing spirit of regulatory permissiveness regarding interconnection are prompting increased interest in the application of STATDM, packet, and line or circuit switching concentration techniques in all types of computer-communication applications. This chapter attempts to present a balanced perspective of how multiplexing and concentration relate both to the end user and to the common carrier. To be sure, this is a difficult objective, particularly in light of today's increasingly nebulous distinction between the once well-separated roles of carrier and user.

7.1. MULTIPLEXING AND CONCENTRATION TECHNIQUES CONTRASTED

The motivations for line sharing stem from economies of scale in the cost of bandwidth and from the increased channel utilizations such devices can produce when serving a large terminal population with predominantly unscheduled requests for service. For example, with today's domestic tariff structure, leased voice-grade lines typically cost up to twice as much as low speed lines of the same length. However, such voice-grade facilities are generally capable of transmitting data at speeds at least 20 to 3[0] times higher than those of the typical low speed link. Thus, by using line sharing, the cost per unit of capacity (bits per second) in a fully utilized voice-grade line can often be reduced to less than one tenth of that of an equal-length low speed line. These economies of scale in the cost of bandwidth generally extend over to carrier-provided broadband links as well.

Before proceeding further, it is appropriate to distinguish between multiplexing and concentration. *Multiplexing* generally refers to static channel derivation schemes in which given frequency bands or time slots on a shared channel are assigned on a fixed, predetermined (a priori) basis. Thus a multiplexer has generally balaced input and output bit rat

capacities. *Concentration,* by contrast, describes schemes in which some number of input ports dynamically share a smaller number of output subchannels on a demand basis. Concentration thus involves a traffic smoothing effect not characteristic of multiplexing. Since the aggregate input bit rate and output bit rate need not be matched in a concentrator, statistics and queuing play important roles. Of the techniques discussed in this chapter, FDM and STDM are examples of multiplexing. Message switching, packet switching, and line switching illustrate the concept of concentration, whereas STATDM is effectively a hybrid sharing scheme embodying salient features of both methods. For this reason, a statistical time-division multiplexer is sometimes called a dynamic multiplexer or a multiplexer-concentrator [1,2]. In such systems, subchannels have a statistically high probability of being available for a given input port, but this availability is not a certainty, as would be the case with STDM.

7.2. FREQUENCY-DIVISION MULTIPLEXING (FDM)

Frequency-division multiplexing partitions a limited-bandwidth communication channel into a group of independent lower speed channels, each of which utilizes its permanently assigned portion of the total frequency spectrum. As shown in Figure 7.1, each channel in the sharing group thus uses a frequency slot that contains the unique pair of frequencies needed for sending its binary data signals. When FDM is used on a voice-grade line, each subchannel may typically transmit data asynchronously at speeds up to 150 bits/sec, although in special cases at faster rates. One of the limitations of FDM arises from the need for guard bands or safety zones between adjacent subchannels to prevent the electrical over-

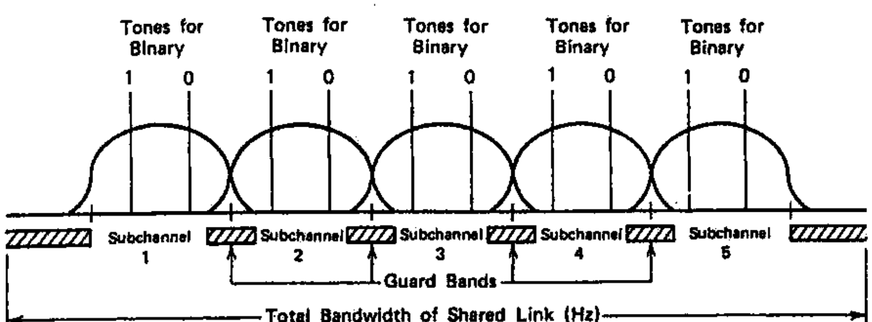


Figure 7.1. Spectrum partitioning and signaling frequency assignments in typical FDM System.

lapping of signals. These guard bands impose a practical limit on the efficiency of an FDM system. For example, with state-of-the-art FDM equipment operating on a leased voice-grade line, the maximum composite or aggregate low speed bit rate achievable will typically range from 1800 to 2000 bits/sec, although in some cases slightly higher. Generally speaking, other types of sharing must be used if a higher aggregate bit-rate requirement exists.

The primary advantage of FDM to end users is its relatively low cost in applications where its aggregate bit-rate limit is not constraining. Some of this economy is provided by eliminating the need for a separate modem or data set at each remote terminal site, since the FDM device is usually designed so that it also performs the modulation and demodulation functions. Also, FDMs are readily cascadable or, in other words, have features that facilitate dropping and inserting at intermediate points along a multiplexed channel. Thus FDM is particularly cost effective in multiplexing an unclustered terminal group (like the one shown in Figure 7.2) whose aggregate bit rate does not exceed the limit mentioned above.

As shown in Figure 7.2, each FDM subchannel is connected to the communications controller with a separate port. Viewed by the network control software, an FDM line containing the three subchannels and three terminals shown in Figure 7.2 cannot be distinguished from a configuration employing three separate leased lines. Thus a user may employ FDM to combine traffic of different terminals onto one communication line without using polling software.

In cases where a user is willing to control the remote terminals of a network with some type of polling software (or let multiple remote terminals share a port on the communications controller), a second level of

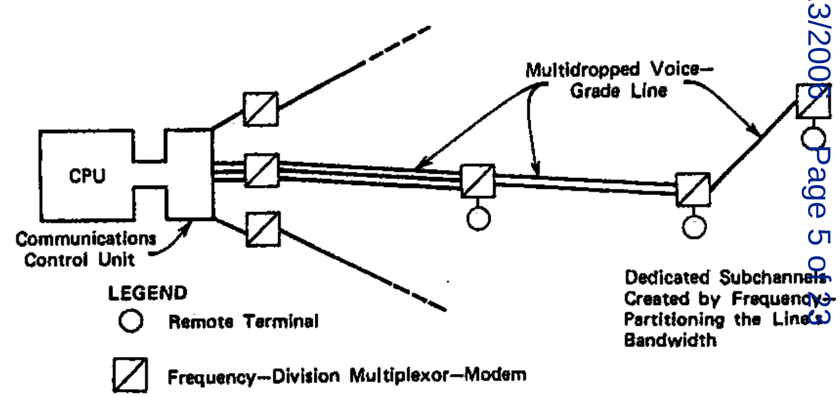


Figure 7.2. The Use of FDM to service unclustered terminals.

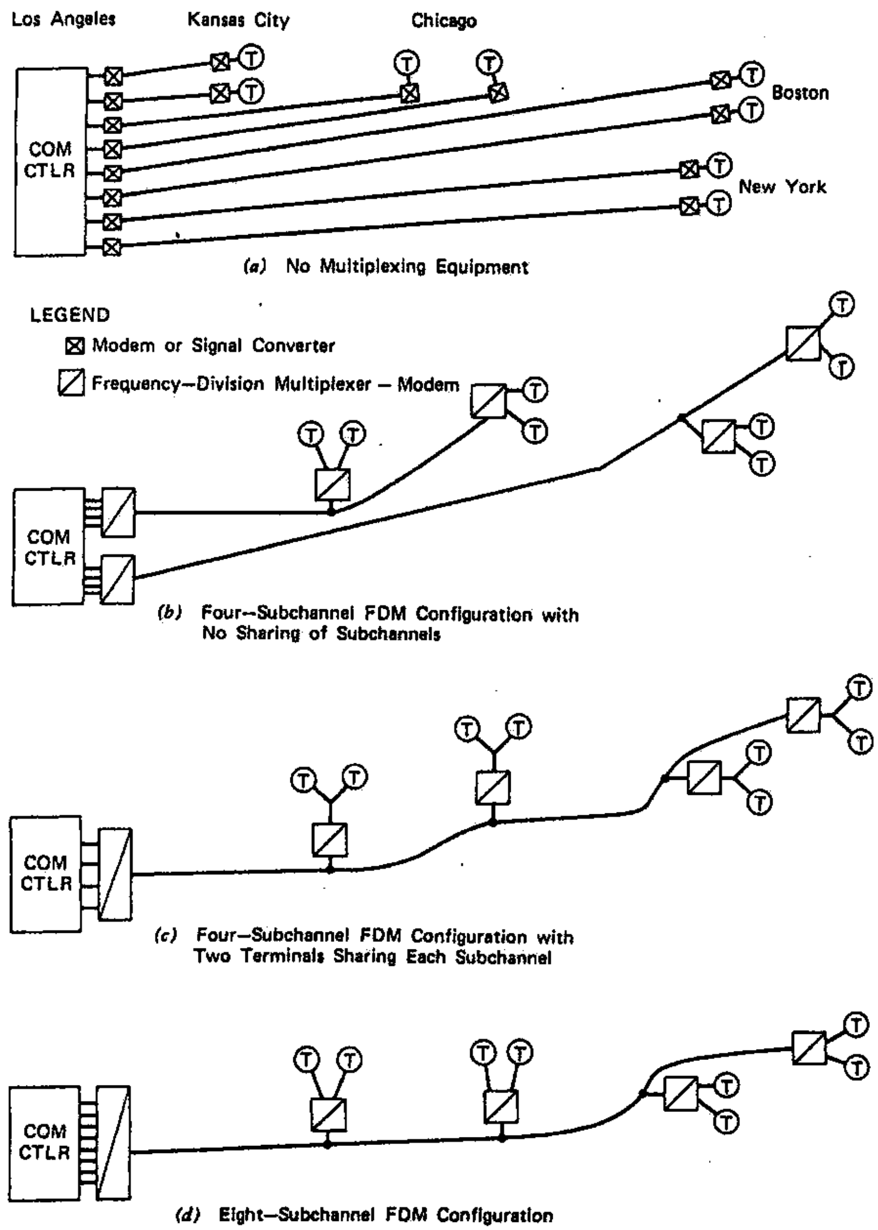


Figure 7.3. Alternative FDM configurations for connecting eight terminals into CPU, using voice-grade lines.

sharing becomes possible. The capacity of the voice-grade line is shared by the FDM equipment, which creates multiple independent subchannels. Each subchannel in turn may be time shared, on either a contention or a polled basis, by multiple remote terminals.

For example, imagine that two asynchronous terminals in each of the following cities—Boston, New York, Chicago, and Kansas City—are to be tied to a central computer in Los Angeles. Figure 7.3 illustrates four possible configurations, the first of which uses individual leased lines to each remote terminal. The second approach utilizes FDM equipment with an assumed capacity of four subchannels per voice-grade line and no sharing of the subchannels. The third approach, which clearly has a still lower line cost, uses FDM equipment with an assumed capacity of four subchannels per voice-grade line and two terminals per FDM subchannel. The fourth approach has the same line costs as the third configuration. However, an FDM system with an assumed capacity of eight subchannels per voice-grade line (and no sharing of subchannels) is postulated. Note that all configurations (except the one in which sharing of the subchannels is permitted) require eight ports at the central site. Only four ports are required when each subchannel can be shared by two remote terminals.

Other popular examples of the application of frequency-division multiplexing techniques are their use in special modems to create a full-duplex channel over a two-wire circuit (discussed in Chapter 2) and to provide extra low-speed teletype-grade channels on voice-grade circuits, primarily in international applications. In such situations, the collective costs of separate voice-grade and teletype (sub-voice-grade) lines are often sufficiently large that it may be less expensive to operate one leased voice-grade line (for either voice communications or data transmission up to 9600 bits/sec) concurrently with one or more independent low speed side channels on the same physical line. The analog data modems can usually be switched out in favor of a telephone at each end, enabling either voice or data transmission to take place independently of slow speed subchannel activity.

7.3. SYNCHRONOUS TIME-DIVISION MULTIPLEXING (STDM)

Time-division multiplexing devices that create a permanently dedicated time slot or subchannel for each port in the sharing group are classified as STDMs. By contrast, statistical or asynchronous TDMs dynamically allocate the subchannels or time slots on a statistical basis to increase line

efficiency by providing time slots only for ports actively transmitting data.[3]

As shown in Figure 7.4, STDMs share a synchronous communication line by cyclically scanning incoming data from input ports, peeling off bits or characters, and interleaving them into frames on a single high speed data stream. This effect is similar to that of a high speed conveyor belt picking up objects arriving at a common point from several lower speed belts. In utilizing a given channel, STDM is generally more efficient than FDM since it is capable of using the entire bandwidth available.

For example, STDMs can generally operate over dedicated voice-grade lines at speeds of 4800, 7200, and 9600 bits/sec, whereas FDMs' practical limit on the same line is probably in the 2000 bits/sec speed range. Generally speaking, the multiplexed data stream is transmitted serially, bit-by-bit, at a rate governed by the circuit-signal converter combination. The split-stream modems introduced in Chapter 2 are popoular devices that combine the modem and STDM functions into the modem device when the input port speeds are integer multiples of 2400 bits/sec and the total input rate does not exceed 9600 bits/sec.

Although voice-grade lines are shared in the majority of current STDM applications, a recent regulatory development now enables end users to multiplex broadband carrier links with customer-provided STDM equipment. Whereas STDMs can multiplex traffic from either asynchronous (start/stop) terminals, other synchronous devices, or combinations thereof, FDMs are generally used to multiplex only asynchronous terminals, although this is not an intrinsic limitation.

Contemporary STDMs may perform either bit or character interleaving on the shared line when serving start/stop terminals exclusively. In these applications, character interleaving is usually more efficient since a modest amount of bandwidth compression is possible. The start and stop bits of each character entering the STDM may be stripped off before the character's insertion into the frame of multiplexed data. (Figure 7.5 illustrates the technique of character-interleaved STDM, including data characters and the encodings of various end-to-end control signals.) Any bits stripped from incoming data characters are reinserted by the demultiplexing unit before distribution of the characters to their respective

[3]The reader should not confuse asynchronous time-division multiplexers, in the sense of their current interpretation, with STDMs that multiplex asynchronous (start/stop) terminal devices. Throughout this chapter, "STDM" describes the time-division technique in which dedicated subchannels are created for start/stop devices, synchronous devices, or a combination thereof. "Statistical" or "asynchronous TDM," by contrast, describes all schemes where the multiplexer creates subchannels dynamically, regardless of the type of device being multiplexed.

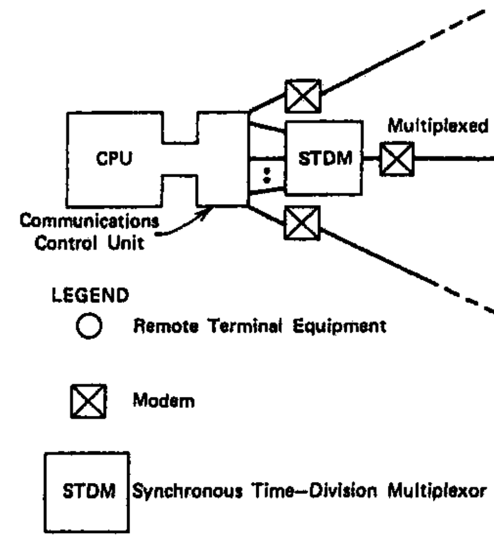


Figure 7.4. Synchronous TDMs in a teleprocessing network.

output ports. Thus the incoming characters are effectively encoded using fewer bits per character for transmission over the shared link, thereby enabling an aggregate low speed bit rate of 1.1 times the shared link's transmission rate to be accommodated in typical situations.

In newer applications involving the use of STDMs to multiplex synchronous data streams, the STDMs generally employ bit interleaving,

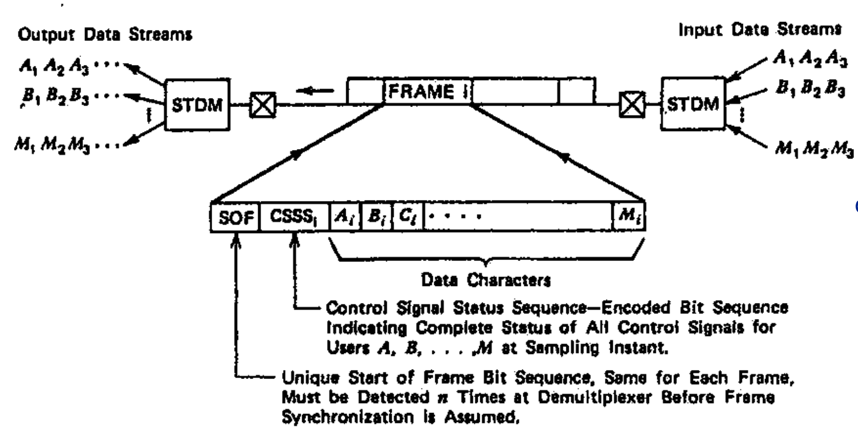


Figure 7.5. Character-Interleaved STDM and frame format.

disregarding the textual content of the incoming data streams. This data transparency is an important requirement for multiplexers being incorporated in the long haul trunks of synchronous digital data networks now being implemented by certain users and common carriers. Whether bit or character interleaving is used, special predetermined code sequences are utilized between STDMs to define the beginning of each new frame of multiplexed data. Demultiplexing is thus accomplished by the assumption of an implicit relationship between the output line or buffer address and the relative position of the time slots in an arriving frame.

When all the multiplexed terminal devices are unbuffered, the problem of fixing the scanning rates within the STDM is straightforward—the subchannel scan rates are matched to the transmission rates of the respective input lines. Ordinarily this speed corresponds, in turn, to the operating speed of the remote terminal device being served. However, when messages and message segments are queued in remote terminal buffers, the solution to the scan rate assignment problem is not so obvious. In References [3] and [4], Doll has developed a queuing theoretic design technique for determining scan rates within the STDM so that the average queuing delay experienced at the remote terminals in the sharing group is minimized.

Noise disturbances on the shared channel can cause a variety of errors, depending on whether character or bit interleaving is used. With character interleaving, an individual data bit error will at worst cause a single output character to be received in error. With bit-interleaved STDM, a similar anomaly could cause the demultiplexing unit to deliver the outputs to the wrong addresses, unless the STDMs contain their own error control capability. As a consequence, character STDMs are less sensitive than bit multiplexers to channel disturbances, although resynchronization (reestablishing the start of a data frame) takes somewhat longer than with bit-interleaved STDMs.

Higher quality STDM devices have been designed on the philosophy that random and burst errors can be allowed to cause data errors, but must virtually never cause errors in the end-to-end control signals between user terminals or in the internal network control signals between STDMs themselves. This goal may be accomplished without substantial degradation of shared link capacity by using highly redundant encodings of all vital control signals. For example, frame synchronization is obviously critical and must be preserved in the presence of noise bursts. Elaborate time averaging and/or thresholding schemes have been devised whereby frame synchronization is assumed at the demultiplexer only after a unique bit sequence has been detected a specified number of times in a given time period. Similarly, frame synchronization is assumed to be lost

only when this same condition cannot be detected. Typical noise distu… bances may thus be smoothed over without catastrophic effect, eve… though some data bit errrors may occur during intervals when fram… synchronization is being reestablished.

In comparison to FDMs, STDMs are expensive to cascade because … relatively complete STDM system must be used at any point where one … more subchannels are being inserted or removed. Also, when STDMs ar… used in cascade, the problem of coordinating the timing across multipl… synchronous links must be addressed. Fundamentally, two choices exi… here—to use one master clock from which all system elements deriv… their timing, or to use independent synchronous clocks on each link. Th… former alternative may be illustrated by the simple configuration shown i… Figure 7.6. The modem at A provides the master clock signal, and th… lefthand modem at B derives its clock from the incoming data on link AB… The righthand modem at B is slaved to its lefthand counterpart at B and, i… turn, provides a master clock to link BC. With the alternative approach … independent synchronous clocks, some type of elastic buffer must b… provided at node B to absorb data buildups caused by slight variations i… the rates of the two independent clocks. Most STDM networks to dat… have been implemented using the single-master-clock scheme for variou… economic and reliability reasons.

As with FDM, an STDM configuration provides each port in the sharin… group with its own dedicated appearance at the communications contr… unit. (See Figure 7.4.) Here it is assumed that STDMs are used in pair… one for multiplexing and the other for demultiplexing. It is possible t… eliminate the central site STDM if the communications control unit ca… perform the STDM function in hardware or software. (An early IB… Corporation STDM product combination known as the 2712 Multiplex… operated with a hard-wired transmission control known as the 270… eliminating the need for two separate central site devices.)

Since most computer vendors have not emphasized time-division mu…

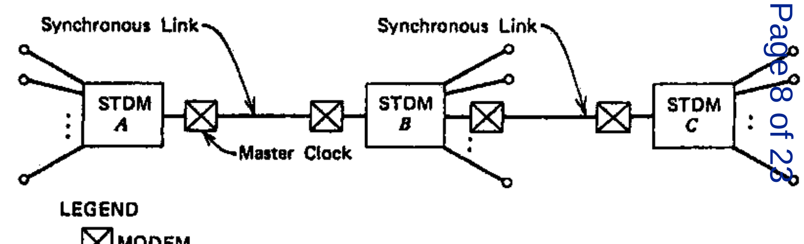


Figure 7.6. A synchronous TDM network.

tiplexing equipment (and vice versa), users continue frequently to use the approach of paired STDMs for multiplexing and demultiplexing. It creates a well-defined hardware interface point between the communications controller and the network. It also enables conventional, vendor-supported teleprocessing control software to be utilized without modification or extension. The advantages arising from these two characteristics often outweigh the extra costs associated with the two-separate-box approach to time-division multiplexing.

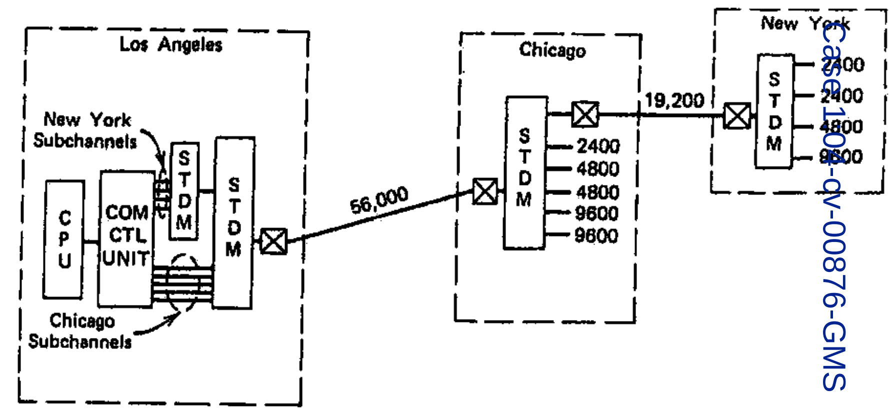
Subsequent discussions of statistical multiplexing, packet switching, and intelligent communications networks based on minicomputers will reveal certain advantages of dynamic bandwidth sharing not available with conventional STDM. When such benefits can be coupled into networks with minimal requirements for modifications to the line control software, the user stands to achieve the best of both worlds—a flexible network and the full support of vendors supplying the teleprocessing control software.

7.3.1. Configuration Options in Cascaded STDM Networks

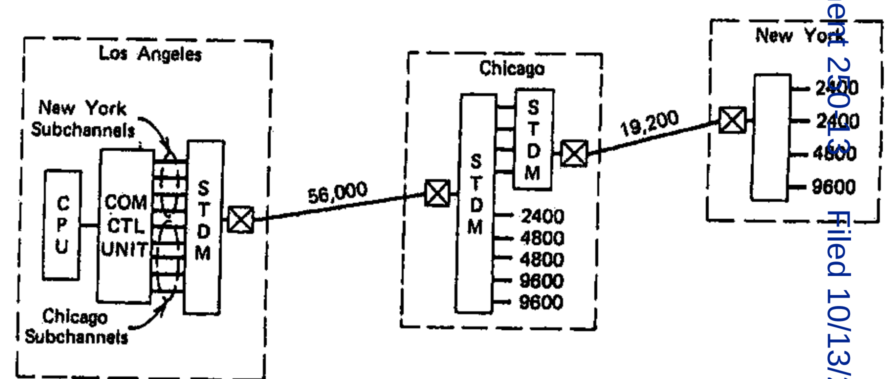
Consider a user with a Los Angeles CPU and numerous remote terminals in Chicago and New York. For illustration purposes, assume that the following terminal–speed combinations are required:

New York	Chicago
Two 2400 bits/sec terminals	One 2400 bits/sec terminal
One 4800 bits/sec terminal	Two 4800 bits/sec terminals
One 9600 bits/sec terminal	Two 9600 bits/sec terminals

Also assume that 19,200 and 56,000 bits/sec line speeds are available for use on the multiplexed STDM links; the problem is to connect all nine remote terminals into separate ports on the Los Angeles communications control unit. Figure 7.7 illustrates that the New York channels may be demultiplexed in either Chicago or Los Angeles. If demultiplexed in Chicago, the New York channels must be remultiplexed with the Chicago traffic onto the Chicago–Los Angeles link. This so-called drop-and-insert arrangement would provide convenient access to the New York channels in the Chicago site if such an arrangement is desirable. Also, extra STDM capacity could be used between New York and Chicago, if required. On the other hand, the Los Angeles demultiplex alternative would place more equipment in one location, providing easier access for maintenance and diagnostic functions from a centralized location.



(a) Completely Centralized Demultiplex Approach



(b) Drop and Insert Approach, Where New York Subchannels Are Available in Chicago

Figure 7.7. Configuration options in a cascaded STDM network.

The user planning a cascaded STDM network needs to consider carefully whether the flow patterns in his network are centralized or noncentralized, and how the initial STDM network layout could be affected by a need to add new channels at a future date. The strategy of performing as much demultiplexing of inbound data streams as possible at the central site appears to offer numerous advantages in centralized-flow situations. On the other hand, the drop-and-insert scheme of Figure 7.7(*b*) affords more flexibility in decentralized applications where the subchannels need to be directly accessible at intermediate locations in the STDM cascade.

7.3.2. Similarities and Differences between FDM and STDM

Both FDM and STDM are widely used for reducing costs in end-user networks, the cost-reduction possibilities in either case arising from often-present economies of scale in the cost of bandwidth. Meaningful cost comparisons of various multiplexing techniques obviously require that necessary mileage-independent costs such as those for line terminations, multiplexers, and signal converters also be included. However, many present tarriffs are structured so that multiplexing can produce substantial net savings, even after the costs of all required equipment are factored into the overall comparison. When the aggregate low speed bit rate for all terminals does not exceed 2000 bits/sec (give or take 10%), either FDM or STDM can probably be used; however, with current technology, FDM will probably be more cost effective than STDM, particularly when remote terminals are not clustered at a single site. Whenever a higher aggregate bit rate is required or any synchronous terminals are included in the sharing group, STDM will usually be dictated. However, in higher bit rate applications involving geographically dispersed terminal locations, an integrated blend of FDM and STDM will be the most sensible choice. Frequency-division multiplexing can span isolated terminal sites, creating traffic clusters that are then synchronously multiplexed into one or more computer sites.

Historically, the predominant usage of multiplexing has involved the derivation of low speed teletypewriter-grade channels on voice-grade lines. More recently, newer applications of STDM have appeared, particularly with the increased availability of higher speed synchronous modems, the initial availability of all digital data networks from conventional carriers, the entry of specialized carriers into the data network business, and recent provisions enabling customer-provided multiplexing equipment to be used over carrier-provided wideband links. If the costs of a multiplexed wideband link between two points can initially be justified, users may expect generally lower error rates on all derived voice-grade and low speed channels, substantially increased flexibility, and the opportunity to assign initially unused capacity at a later date without increased modem or line costs on the shared link.

These points are now illustrated in detail using specific examples. The reader is cautioned that the tariffs used in these examples are strictly for illustration purposes. Exact rates should always be obtained from the carrier. Tariffs used were in effect at the time when comparisons were made.

Example Problem 1: Comparison of FDM and TDM

Assume that a central computer located in downtown Chicago needs to provide 712 mile connections to 10 separate terminals in the same building in New York City. The terminals operate at 110 bits/sec. Determine whether individual leased lines, frequency-division multiplexed analog lines, or time-division multiplexed analog lines would be the most cost-effective networking strategy. The cost assumptions below are illustrative of typical industry prices for comparable equipment at publication time.

Cost Assumptions:

Cost of FDM equipment:	\$30 per month per channel end
Cost of TDM equipment:	\$250 per month (fixed) plus \$20 per month per low speed port
Cost of modem equipment:	\$50 per month for 2400 bits/sec units
	\$100 per month for 4800 bits/sec units
	\$200 per month for 9600 bits/sec units

Cost of individual low speed lines—AT&T Series 1006 FDX
Cost of voice-grade lines—AT&T Series 3002 (MPL)

Option 1: Individual Low Speed Lines

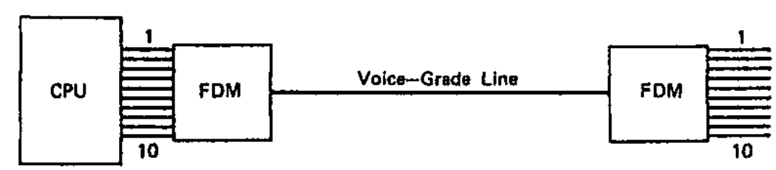
Monthly Cost of Individual 110 bit/sec circuit, including signal conversion equipment

$$= \underbrace{2.023 \times 100 + 1.416 \times 150 + .811 \times 250 + .605 \times 212}_{\text{Mileage}}$$

$$+ \underbrace{2(36.15 + 14.45)}_{\text{Terminations}}$$

So the monthly cost of 10 circuits provided with individual lines = \$8469.10.

Option 2: Frequency-Division Multiplexing Approach

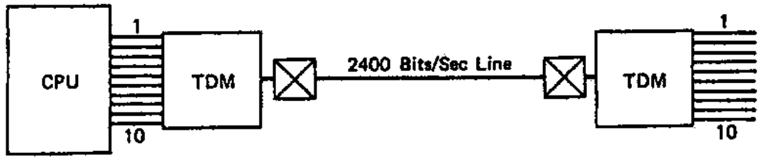


Since there are 10 FDM channel ends at each end of the voice-grade line, the FDM equipment cost is 20 × 30 = \$600/month. The monthly cost of the line (assuming no special conditioning) is

$$175.20 + 0.66(712 - 100) + 2 \times 25 = \$629.12$$

Hence the total cost of the FDM approach is \$1229.12. This is clearly more attractive than Option 1.

Option 3: Time-Division Multiplexing Approach



Monthly TDM costs are 2[250 + 10 × 20]	=	\$900
Monthly modem costs are 2 × 50	=	100
Monthly line cost is (from above)	=	629.12
TOTAL COST		\$1629.12

Hence the FDM approach is the most attractive of the three considered.

The reader should also consider other types of approaches such as WATS, dial-up, or the packet switching networks of value-added carriers before choosing a specific configuration. However, these options would require some idea of traffic volumes and usage patterns.

Example Problem 2: Another Comparison of FDM and TDM

Assume the same problem as for Example 1 except that the number of terminals in New York is increased from 10 to 20. Assume that an individual FDM system can derive a maximum of 12 subchannels at 110 bits/sec over one voice-grade line and that the TDM system can provide 20 subchannels on a 2400 bits/sec line, 40 subchannels on a 4800 bits/sec line, and 80 subchannels of 110 bits/sec on a 9600 bits/sec line.

Option 1:

$$\text{Monthly cost} = 20 \times 840.50 = \$16{,}810$$

Option 2:

$$\text{Monthly cost} = 2 \times 1229.12 = \$2458.24$$

(since two separate lines and four FDM devices are now required).

Option 3:

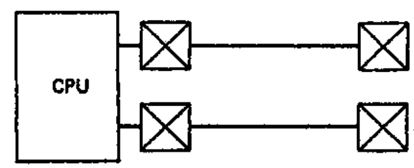
Monthly TDM cost = 2[250 + (20 × 20)]	=	1300.00
Monthly modem cost = 2 × 50	=	100.00
Monthly line cost	=	629.12
TOTAL COST		\$2029.12

Now the TDM approach is more favorable because of the require number of channels.

Example Problem 3: Synchronous TDM Possibilities

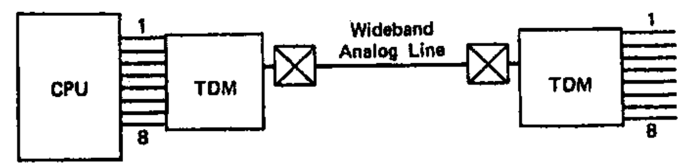
Assume that a Chicago computer center requires eight ports of 480 bits/sec for connections of different synchronous terminals in a New Yor City regional office. Find the best way to provide the service, assumin that the following alternatives are available: (*a*) individual analog voic lines with modems, (*b*) wideband analog lines, and (*c*) digital service suc as DDS (see Chapter 3 for service explanation and cost assmptions). Use the same equipment costs except that the cost of a synchronous TDM input port is assumed to be \$40/month.

Option 1: Individual Analog Lines



Monthly cost = 8 × (629.12 + 200) = \$6632.96

Option 2: TDM over Wideband Analog Lines



Monthly line cost of AT&T Series 8000 wideband analog lines, includi signal conversion equipment,

$$= \underbrace{16.20 \times 250 + 11.40 \times 250 + 8.15 \times 212}_{\text{Mileage}} \times \underbrace{2 \times 460}_{\text{Terminations}} = \$9547.80$$

Monthly TDM cost = 2[250 + (8 × 40)] = $1140
TOTAL COST = $10,687.80

Option 3: Individual Dataphone Digital Service (DDS) Lines at 4800 bits/sec

Monthly cost of individual 4800 bit/sec DDS line

$$= \underbrace{0.26 \times 71}_{\text{Mileage}} + \underbrace{2 \times 20.60}_{\substack{\text{Intercity}\\ \text{link}\\ \text{termination}}} + \underbrace{2 \times 87.55}_{\substack{\text{Local}\\ \text{access}\\ \text{line}}} + \underbrace{2 \times 15.45}_{\substack{\text{Signal}\\ \text{conversion}}} = 688.64$$

Monthly cost of eight separate lines = $5509.12

Option 4: Multiplexed 56,000 bits/sec Dataphone Digital Service

Monthly line cost, including signal conversion equipment,
= 4.12 × 712 + 2 × 64.50 + 2 × 206 + 2 × 20.60 = $3515.64
Monthly TDM cost (from above) = $1140
TOTAL COST = $4655.64

Summary of Example

The best solution for this example appears to be multiplexed 56,000 bits/sec Dataphone Digital Service. However, the monthly cost savings over individual 4800 bits/sec DDS lines needs to be weighed against the fact that the reliability properties of the multiplexed network are much poorer. A failure of the TDM equipment or of the multiplexed line would cause all channels to become unavailable. The operational costs of such a catastrophic failure situation may mean that the multiplexed network is less desirable, in spite of the cost savings it offers.

The reader is also cautioned against generalizing about the relative merits of various networking approaches from this example. The conclusions here, and for other examples as well, are strongly dependent on all the cost assumptions, distances, and required numbers of channels. However, the solution techniques are quite general and are equally useful for alternative tariff structures and equipment costs.

7.4. STATISTICAL TIME-DIVISION MULTIPLEXING (STATDM)

Much of the theoretical work relating to STATDM has been done by Chu [2,5–8]. Related contributions have been made by Rudin [20], Birdsall et

al. [9], Pan [10], Gordon et al. [11], and Chang [12]. Reference [30] als discusses STATDM in detail. Statistical time-division multiplexing differ from STDM in that a dedicated subchannel is not provided for each port i the sharing group. Since, under certain conditions of heavy loading, STATDM may be incapable of accommodating all the terminals in it sharing group, statistics and queuing become important considerations Thus it is a hybrid form of multiplexing and concentration.

The fundamental idea of STATDM is to exploit the property of STDM systems that many of the time slots in the fixed-format frames are waste because a typical sending terminal will actually be transmitting data les than 10% of the time it is communicating with the CPU. A more detaile discussion of typical traffic arrival statistics is presented by Fuchs an Jackson [13]. As shown in Figure 7.8, STATDM dynamically allocates th time slots in a frame of data to the currently active users, reducing th fraction of wasted time slots and thereby increasing overall line utilizatio and throughput.

Although the diagram of Figure 7.8 illustrates addressing informatio being transmitted with data in each slot, it is of course not necessary to d so in cases where such a procedure could lead to excessive overhead. A alternative would be to send demultiplexing address information in control frame only once at the beginning of each dynamic subchann establishment. This demultiplexing rule can be dynamically updated onl when subchannels are added or removed, without the need to incluc address information bits explicitly with the data in each slot. Anoth possibility is to vary the slot widths for the individual ports or to encode control signal that tells the demultiplexer exactly which ports are idle in given frame.

Most estimates of the exact performance improvements attainable wi STATDM over STDM have to date been based on analytical studi described in certain of the references previously cited. From Chu's an lytical studies, it would appear that from two to four times as many use could be accommodated on a voice-grade line as with STDM, assumi an application environment where either method could be used. In certa situations where low duty cycle terminals are serviced by a statistic multiplexer over a broadband link, the margin could be substantial greater.

The tradeoff disadvantages of statistical multiplexing are the costs substantially more elaborate addressing and control circuitry, the need f data buffers to hold incoming messages, and the possibility of blocki and/or appreciable queuing delays under heavily loaded conditions. T references previously noted contain substantial traffic studies, investig ing the relationships between such factors as traffic intensity, distrib

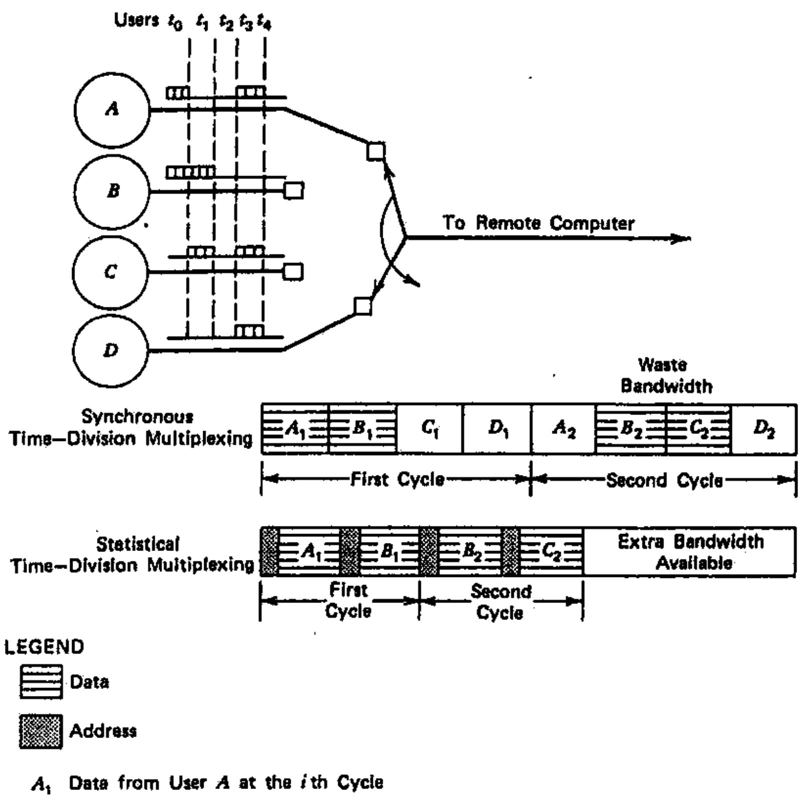


Figure 7.8. STDM contrasted with STATDM.

tions of message arrivals and lengths, queue sizes, queuing discipline, and blocking probability. To illustrate, several of the major results described in Chu [2,5–8] are now summarized.

Chu has analyzed a Markov model of a statistical multiplexer in which messages arrive at a finite-capacity multiplexer according to a batched Poisson process where the size of the batch corresponds to the length of the arriving message in characters. A unit service interval μ is the time to transmit a character on the shared line; for a synchronous line with a transmission speed of R characters/sec. $\mu = 1/R$. Message lengths are assumed to be geometrically distributed with mean $\bar{l}$, and the number of messages arriving during a unit service interval is Poisson distributed with a mean rate of λ messages every 1 sec. The buffer overflow probability is obtained from the steady-state solution to the state equations for an embedded Markov chain. The average queuing delay per message D is shown to be given by

$$D = \frac{\lambda(2-\theta)}{2(\theta-\lambda)\theta} \qquad \text{(character service times)}$$

where $\theta = 1/\bar{l}$. The buffer overflow probability is assumed to be sufficiently small that virtually all traffic arriving at the multiplexer is transmitted over the line.

At the demultiplexing end of the line, Chu has used a simulation model to estimate overflow probabilities and the following analytic relationship to describe average waiting time W_i for sending messages to the ith destination:

$$W_i = \frac{\rho_i(2\bar{l}_i - 1)}{2(1-\rho_i)} \qquad \text{(character service times)}$$

where $\bar{l}_i$ = average message length for the ith destination
λ_i = message arrival rate for the ith destination
μ_i = transmission rate for the ith destination
$\rho_i = \lambda_i \bar{l}_i / \mu$

Figure 7.9 indicates the parameters of a generalized model that may be used to evaluate design tradeoffs in configuring statistical multiplexers. It also suggests that a statistical multiplexer can perform two levels of concentration when the input terminals are not permanently connected to the input ports.

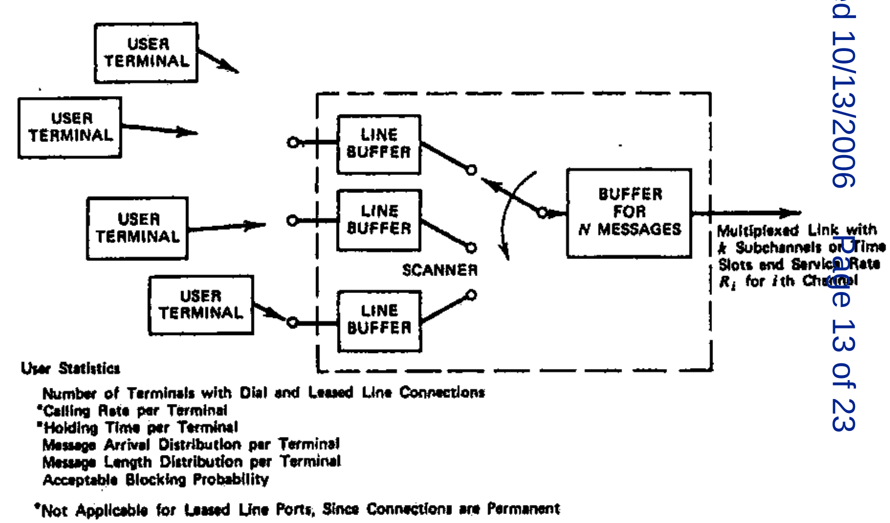


Figure 7.9. Schematic diagram of STATDM for traffic studies.

It appears that STATDM has a promising future, particularly in applications where queuing delays are not of material concern or can be readily minimized. It is this author's conviction that statistical multiplexers will be of primary use, not in replacing STDM en masse, but in new

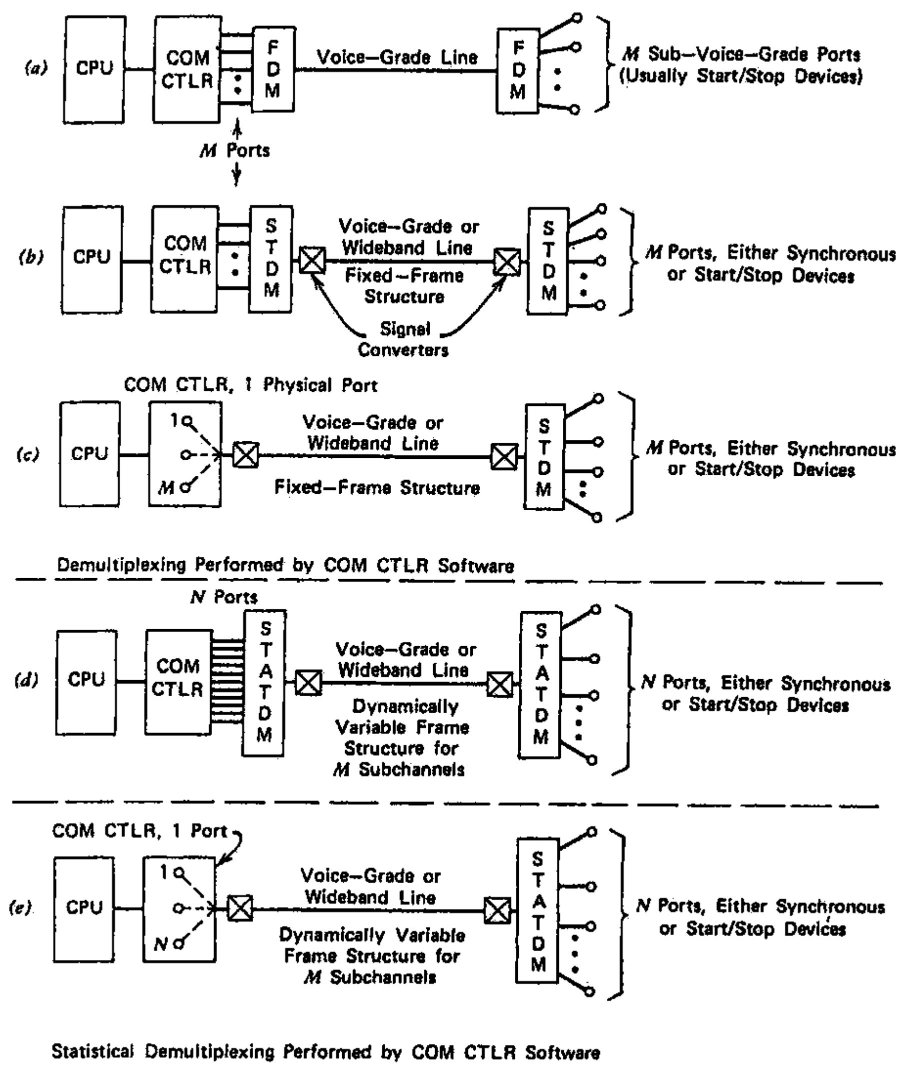


Figure 7.10. FDM, STDM, and STATDM approaches to line sharing. In the STATDM approaches, the number of physical ports, N, connected to the sharing devices is usually larger than M, the number of subchannels created at any instant in time. M changes with time because of fluctuations in user traffic patterns.

applications involving store-and-forward message switching, loop trans mission systems, system-provided error control, and so on. For example CRT display controllers can be equipped to implement STATDM i conjunction with ARQ error control. A request for retransmission woul be issued either when a data block error is detected at the receivin terminal or when the buffer area in the multiplexing control unit is full One of the major problems is incorporating enough intelligence i STATDM to anticipate when an input line is about to become active s that proper steps may be taken to assign the next available time slots t the user in question. A related problem lies in accurately sensing when terminal has completed its transmission so that time slots are not fille with "idle" characters. In certain applications this problem can be miti gated somewhat by having the STATDM unit track the input buffers fo special end-of-message or end-of-block characters. However, this ap proach requires a knowledge of the terminal code format and would be o limited use in transparent text applications.

Figure 7.10 illustrates the relative equipment requirements for usin FDM, STDM, and STATDM to service a remote cluster of terminals Note that demultiplexing of inbound channels may take place either in stand-alone box or in the communications control unit. Because th STATDM approach requires user traffic flows to be monitored anyway the combined-function, single-box approach makes the most conceptua and economic sense. The only potential problem with the combined function approach is the issue of communications control unit horse power. Since STATDM requires virtually continuous tracking of th ports, an already heavily loaded communications control unit may not b able to accommodate the added STATDM function. Even with contem porary microprocessor technology, a separate STATDM unit may ofte be necessary at the central site.

Another substantial advantage of STATDM over STDM is its flexibility in providing different subchannel mixtures across the shared link at different times. This benefit is essentially independent of the increases i throughput resulting from statistical use of shared lines. For example, a STATDM system could easily function as an STATDM for a while an then convert automatically to operate as a conventional STDM system with different subchannel mixtures at other times.

7.5. MESSAGE AND PACKET SWITCHING CONCENTRATION

Message switching concentration (MSC) and packet switching concentra tion are functional extensions of statistical multiplexing involving th "multiplexing" of entire messages and fixed-length portions of long mes

techniques since a buffer queue stores entire blocks of data. Thus, to illustrate MSC, Figure 7.9 would be changed only to reflect the format of data frames transmitted over the shared link. The MSC accumulates message blocks in its buffer until one is completely assembled and the high speed line is available to transmit it. Thus the high speed line transmits variable-length frames of data with appropriate addressing and control information; all data characters in each frame are generally associated with the same source–sink terminal pair.

In most teleprocessing systems where remote concentration is suitable, messages from different users will fluctuate in length. Occasional long messages can occupy the line for prolonged periods, thereby causing other users to experience significant delays while their messages wait in the high speed line queue.

In these situations, predictable system behavior can be restored by having the concentrator chop up long user messages into shorter segments known as packets. As shown in Figure 7.11, packets are individually interleaved onto the shared link. At the receiving end of the link, the packets are sorted out and user messages reconstructed within the individual user buffer areas. As packet sizes become shorter and shorter, the performance of a packet switching concentrator begins to resemble that of the statistical multiplexer previously discussed.

Certain ancillary benefits accrue with MSC and packet switching, such as their inherent capability of performing remote line control, code conversion, error checking, and selective routine, and of implementing the most suitable type of error control on the multiplexed link. The primary disadvantages are economic in nature, and relate to the cost of the stored program computer and buffer storage usually required. A further disadvantage of MSC and packet switching relates to reliability characteristics. Since programmable computers are involved, most individual component failures will be catastrophic in the sense that the entire system becomes inoperative. The standard solution to this problem in most existing message switching networks involves the use of duplexed (paired) devices at the concentration nodes, with the redundant unit being automatically switched into operation whenever the primary fails. Although this approach provides the necessary redundancy, the attendant costs often make other line sharing schemes more cost effective in applications involving moderate traffic levels.

In large, multiuser networks, message and packet switching concentration afford certain flexibility and performance advantages over other line sharing schemes. A substantial amount of recent effort has been devoted to the development of models for predicting queuing delays and allocating channel capacity in MSC and packet networks. Some of the more sig-

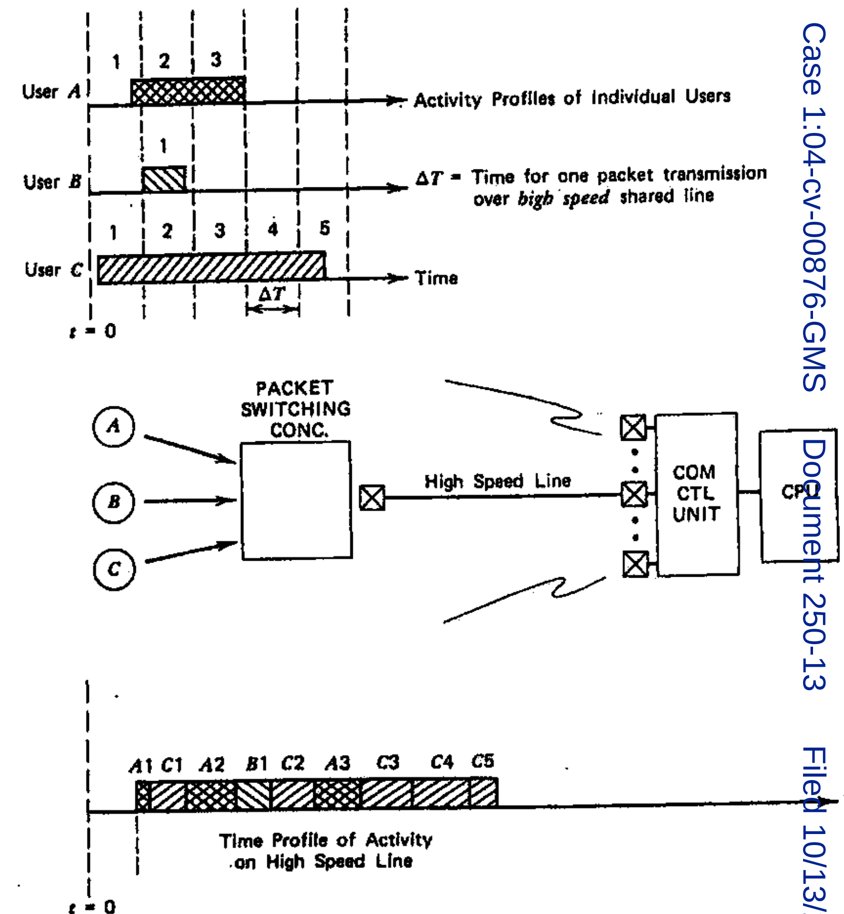


Figure 7.11. Schematic diagram of packet switching concentration

nificant accomplishments have been made by Kleinrock [21,22]; Meist Muller, and Rudin [23]; and Frank et al. [17]. For example, Kleinrc showed that the average message delay in a message switching netwc could be determined using a weighted average of the delays over all t channels of the network, given Poisson message arrivals, exponent message lengths, and other assumptions discussed in the referenc Symbolically stated, the total average delay T is given by

$$T = \frac{1}{\gamma} \sum \lambda_i T_i \quad (7$$

where γ is the sum of all external message arrival rates, λ_i is the average message arrival rate for the ith channel, and T_i is the average delay (including queuing and service times) on the ith channel of the network. Under the assumptions given,

$$T_i = \frac{1}{\mu_i C_i - \lambda_i} \tag{7.2}$$

where μ_i is the reciprocal of the message lengths on the ith channel and C_i is the capacity of the ith channel. This result was ultimately used by Kleinrock to solve analytically the problem in which a given fixed amount of total channel capacity is available for allocation to the channels of a network and it is desired to assign capacities C_i so as to minimize the average delay T given above.

Meister, Muller, and Rudin later solved the same problem for different performance criteria in which the mean rth power of the average delay is minimized. Their approach used the following objective function, of which (7.1) is clearly a special case:

$$T^{(r)} = \sqrt[r]{\sum \frac{\lambda_i}{\gamma} T_i^{(r)}} \tag{7.3}$$

The primary shortcoming of these and the other models developed to this point in time centers around the traffic assumptions that must be made to obtain convenient analytical results. Progress in the development of more generalized non-Poissonian/exponential models has been continual but slow.

7.6. LINE OR CIRCUIT SWITCHING

Circuit switching concentration involves a switching device that electrically bridges a group of n inputs to a group of m output links on a demand basis (n is typically from three to five times the value of m in commercial applications). Ordinarily, the input links and the output trunks to which they are switched have similar bandwidth and transmission properties. A communication channel is thus formed by the electrical concatenation of the input and output link segments within the switch. Thus no message queuing delays are introduced at the switch once a connection is established and held for the duration of a complete data transmission or voice call. When the connection is no longer needed, the corresponding trunk line is freed and made available for assignment to the next input link desiring a trunk connection. Private automatic branch exchanges

(PABXs) are examples of circuit switches. Although historically th have been used primarily in conventional voice telephony, such devic may function equally well as line switching concentrators for compute communication applications. Devices of this type may be built inexpe sively since special purpose computers and software are not required

From a technology standpoint, the connections between inputs a outputs in a circuit switching concentrator may also be accomplish digitally, using a high speed time-division scanning mechanism. T time-division mechanism samples bits from the incoming lines at t correct scanning rate and moves them to the appropriate output trun without delay in much the same manner as an STDM would perform. T multiple outbound channels can also be created by the formation frames on one or more physical lines leaving the switch, indicating further similarity with conventional STDMs. Clearly, however, this s called TDM/circuit switch differs from a conventional STDM in the sen that the aggregate bit rate of incoming lines can generally be quite diff ent from the aggregate bit rate of outgoing lines.

Figure 7.12 illustrates a typical use of a line switching concentrato This illustration depicts all output trunks connected to the same devi (the communications control unit), but line switching units may a function with a mixture of local terminals and remote circuits on both t input and output sides of the switch. Although not widely available at t time, future line switching units may be expected to accommodate mixture of different (from a bandwidth standpoint) types of input a output links as well. These advanced switching units will still conne input and output links of the same type but will be able to concurrent accommodate different groups of link types, using common contr hardware and software.

In the basic line switching unit illustrated in Figure 7.12, several po sibilities exist for handling requests for connection to the o put trunk lines. One possibility involves the servicing of incoming r quests on a first-come/first-served (FCFS) basis. Connection reques arriving at the switch when all output trunks are occupied receive a bu signal and are permanently lost. The multiserver loss (blocking) mod presented in References [14] and [26] may be used to relate blocki probabilities (percentage of calls rejected), the number of input and outp links, holding times, and arrival rates, assuming that blocked requests a lost and incoming requests arrive at random(are Poisson). A more co plex scheduling arrangement would involve incoming requests bei queued when all trunks are occupied. In this situation, the no-loss queui model discussed in References [14], [26], and elsewhere is an appropria vehicle for conducting detailed traffic studies.

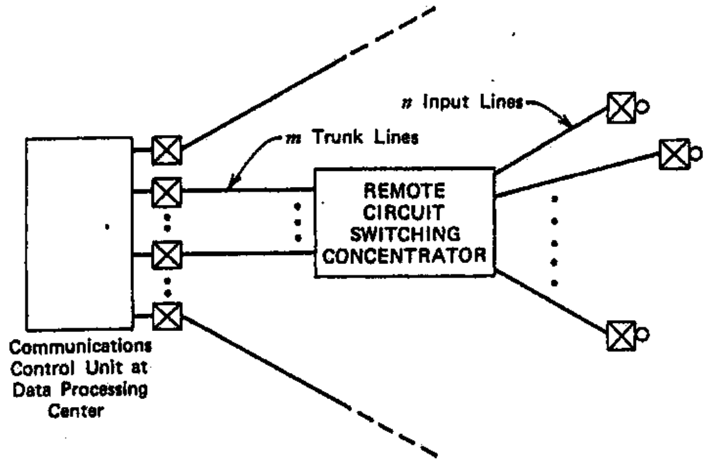


Figure 7.12. Use of circuit switching concentrator.

The application of this multiserver model to a line switching unit that holds incoming calls results in the queuing model shown in Figure 7.13. Here it is assumed that arrivals to the queuing system correspond to call requests which appear at the concentrator. Each call that arrives when one or more of the m identical trunk lines is unoccupied is serviced immediately. Calls arriving at the concentrator when all trunks are busy are queued and subsequently assigned to output trunk lines as the latter become available. Other assumptions for this particular model are that the holding time for all calls is an exponentially distributed random variable

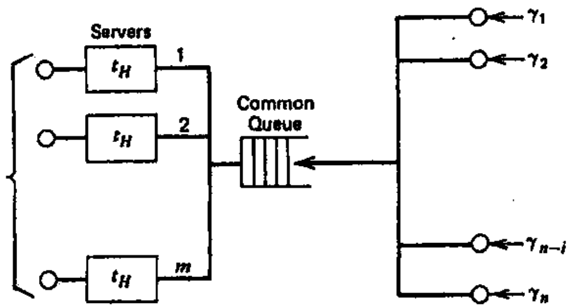


Figure 7.13. Queuing model for line switching concentrator.

with mean $\bar{t}_H$ and that calls arrive at the concentrator according to Poisson process, where γ_i denotes the Poisson average rate of call requests from the ith terminal or source. For data communications applications, the exponential holding time assumption is satisfied by a constant speed trunk line transmitting messages whose lengths are exponentially distributed. Calls correspond to message transmissions, each of which occupies an output trunk line for the duration of a single transmission. Finally, the model assumes that, when a trunk line becomes free after al[l] have been busy, a call is selected from the list of those waiting on a FCFS basis.

The average traffic intensity per trunk line is given by

$$t_H \times \sum_{i=1}^{n} \frac{\gamma_i}{m}$$

and the probability P_N of there being exactly N calls in the system (eithe[r] undergoing service or awaiting access to a trunk line) is given by

$$P_N = \begin{cases} \dfrac{(m\rho)^N P_0}{N!}, & \text{for } N < m \\ \\ \dfrac{(m\rho)^N P_0}{m!\,m^{N-m}}, & \text{for } N \geqslant m \end{cases}$$

where

$$m\rho = \bar{t}_H \sum_{i=1}^{n} \gamma_i$$

and

$$P_0 = \frac{1}{\sum_{K=0}^{m-1} (m\rho)^K/K! + (m\rho)^m/(1-\rho)m!}$$

These relationships may then be used to obtain Q, the average time a ca[ll] request spends both waiting for service and receiving it once a trunk [is] acquired:

$$Q = \bar{t}_H \left[1 - \frac{1}{m(1-\rho)} \times \sum_{k=m}^{\infty} P_k \right].$$

Curves for Q, normalized to the mean holding time t_H as a function of th[e]

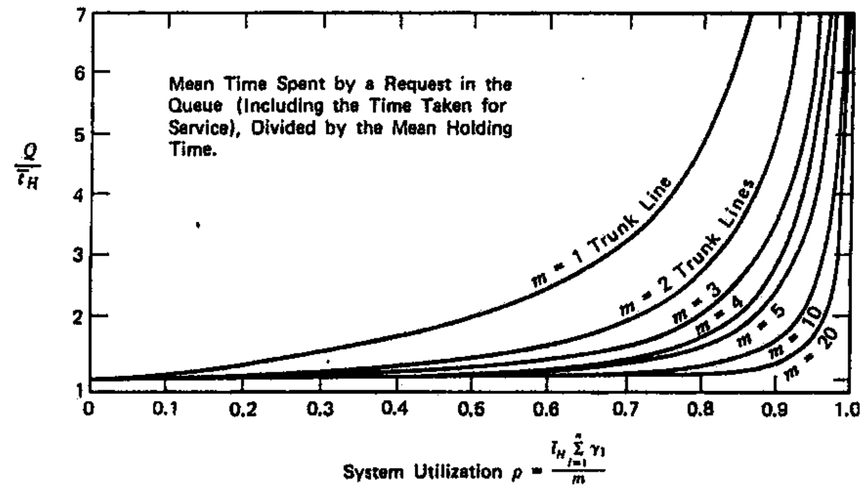


Figure 7.14. Queuing times for accessing and using a trunk line in the line switching concentrator.

number of trunk lines and average system utilization ρ, are plotted in Figure 7.14.

Problems related to the design of networks of circuit switching concentrators are considered in great detail by Benes [24] in his classic book. Similar subjects are also addressed by Rubin and Haller [25]. Since the existing voice telephone network is in effect a very large circuit switching network, the application of line concentrators in computer–communication networks is by no means a new concept. However, they have traditionally been used solely by means of PBXs already installed for voice telephone use. It would appear that special purpose line concentrators designed solely for data transmission applications are likely to become much more popular in end-user networks in the years ahead.

7.7. INVERSE MULTIPLEXING

Recently, substantial interest has developed in the application of multiplexing ideas to create wideband transmission paths using several lower speed lines in parallel.[4] Technically speaking, several independent lines are shared to create one logical path, as shown in Figure 7.15.

[4]Products employing this idea are currently offered by the Codex Corporation, Newton, Massachusetts, and by International Communications Corporation, of Miami, Florida. They are discussed in Reference [27].

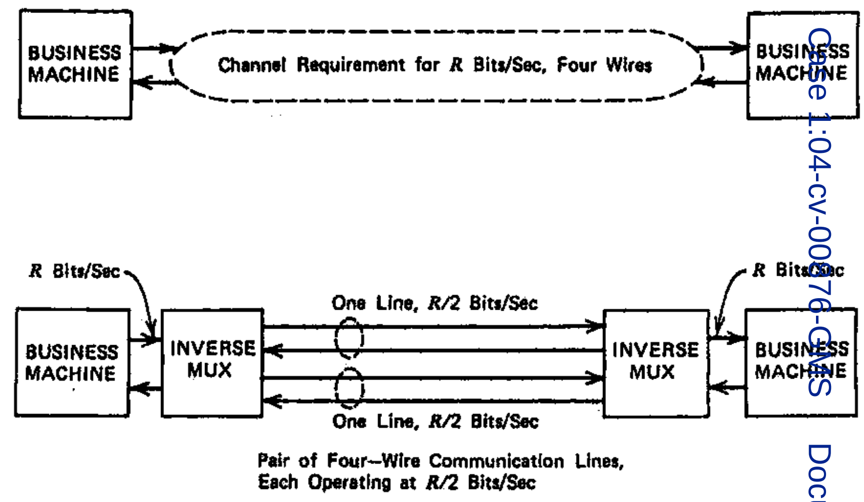


Figure 7.15. Inverse multiplexing example.

The economic justification for such configurations arise from peculiar pricing characteristics in common carrier tariffs, total lack of wideband service availability in some countries, lack of wideband service availability in a timely fashion, and some naturally attractive reliability properties of this inverse multiplexing scheme.

For example, assume that a user requires a 19,200 bits/sec channel between two points. In the United States, a Bell System user would probably be forced to lease a full Series 8000 channel with a maximum bit rate equivalent to 50,000 bits/sec.[5] Since he also must pay the full price of the 50,000 bits/sec link, the arrangement would hardly be economical unless the user could find application for the extra capacity.

An alternative to this wideband offering would be to use a pair of 9600 bits/sec lines connected to these inverse multiplexers at each end. Clearly, four signal converters would be required, two at each end of the individual 9600 bits/sec links. In many applications studied by this author, such a configuration has proved to be extremely economical and cost effective. An attractive reliability feature of the inverse multiplexing technique is the ease of cutting the normal operating speed in half should one of the leased lines fail. Another way of viewing this feature is that the user is able to employ the capacity of both a regular line and a backup line in normal circumstances. Only if one should fail must he cut back to single-line operation. This philosophy is a conceptually and practically

[5]Typical 1976 prices for AT&T Series 8000 channels may be found in Chapter 3.

attractive alternative to the archaic idea of letting a backup leased line stand idle under normal circumstances.

The inverse multiplexing technique could conceptually be extended to provide R bits/sec, using links individually operating at R/N bits/sec, although the application benefits of such a configuration may not be as significant or apparent as in the specific case of two parallel lines (where $N=2$).

7.8. TYPICAL NETWORK CONFIGURATIONS INVOLVING SHARING DEVICES

Having completed the discussion of individual line sharing devices, we now consider some typical network configuration problems involving sharing devices. No new devices are introduced here; rather, the objective is to tie previous material together, using several examples. The five examples presented here involve synchronous time-division multiplexing in four instances and a packet switching concentration situation in the other. For purposes of simplicity and clarity, the time-division multiplexer examples will be illustrated using split-stream modems which combine the STDM and signal conversion function. The concepts noted here are equally valid, however, in situations where stand-alone STDM systems are required instead of split-stream modems.

Case 1. Computer in Los Angeles, 2400 bits/sec terminal in Chicago for remote job entry (RJE), 2400 bits/sec terminal in Chicago for data collection, and 4800 bits/sec terminal in Chicago for inquiry response. Assume that all terminals are on same customer site and that software constraints will not permit different devices to be multipointed on a line.

Solution. For the solution see Figure 7.16. The split-stream modems operating at 9600 bits/sec provide three functionally independent subchannels to the remote terminals. An interesting feature of contemporary split-stream equipment is its ability to be operated in several modes should the multiplexer mixture need to be changed from one time of day to another. For example, the user's inquiry application may not require support after 5:00 P.M. local time in Chicago, whereas the other two terminals need continuing connections. By switching to another mode, either of the following multiplexing possibilities could be achieved: 7200 bits/sec +2400 bits/sec or 4800 bits/sec +4800 bits/sec. Such flexibility should be an important requirement for users planning networks with multiplexing equipment.

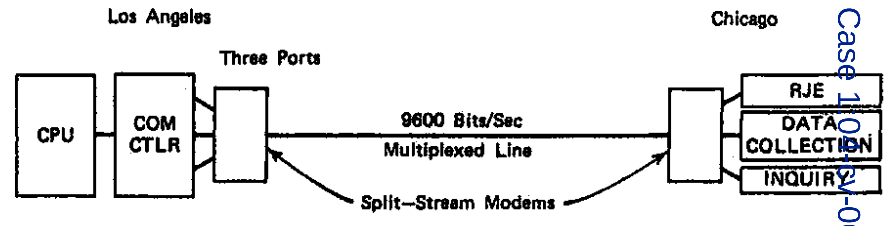
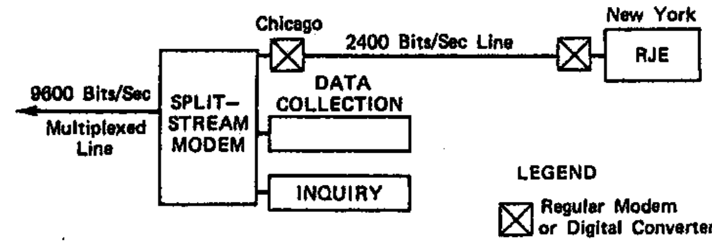


Figure 7.16. Example network for Case 1.

Case 2. Same problem as in Case 1, except that the RJE terminal is moved to New York.

Solution.



Case 3. Same problem as in Case 1, except that one more inquiry terminal in New York is to share the inquiry subchannel with the Chicago inquiry terminal.

Solution. In this configuration of Figure 7.17, it is assumed that the inquiry terminals are polled on a single port from the Los Angeles computer site. The computer views these inquiry terminals as remote drops on a multipoint line. Note that three 4800 bits/sec modems are required, even though the Chicago inquiry station is on the same customer location premises as the split-stream modem. Another alternative, the *channel remoting* arrangement employing the *digital bridge* discussed in Chapter 2, would eliminate the need for one of the remote 4800 bits/sec modems. It is shown in Figure 7.18. Some alternative names used for this digital bridge are *modem sharing unit, modem sharing device, modem contention unit, port contention unit, port sharing unit,* and *port sharing device.*

Case 4. Computer in Los Angeles, 2400 bits/sec RJE terminal in New York, 2400 bits/sec data collection terminal in Chicago, 4800 bits/sec inquiry terminal in Chicago, and 4800 bits/sec inquiry terminal in New

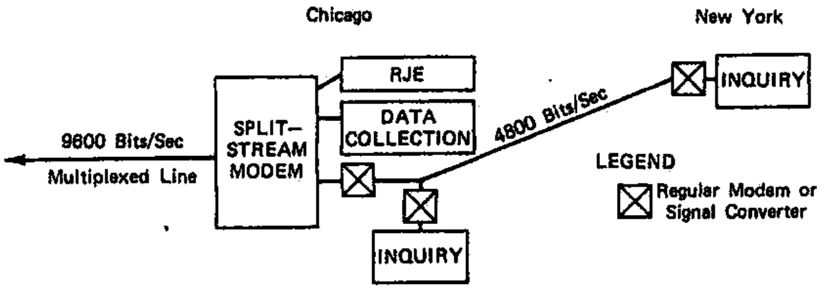


Figure 7.17. Example network for Case 3.

York. Assume that the RJE and data collection terminals cannot be multipointed on the same subchannel because of software restrictions in the main CPU. The inquiry stations may, however, be multipointed.

Solution. See Figure 7.19.

Case 5. Assume that a packet switching concentrator is to be used in Chicago to consolidate numerous kinds of traffic from other locations in the eastern United States. As before, the central computer is in Los Angeles. The example assumes a concentrator with polling capability, and the reduction of all traffic to packets before transmission over the high speed line to Los Angeles. The communications control unit in Los Angeles will collect the packets, sort them out, and deposit them in the correct buffer areas. The illustration of Figure 7.20 shows a broad mixture of line types feeding into the packet switching concentrator. It also illustrates two alternative strategies for handling the second-stage STDM lines. In one case a separate STDM at the concentrator site demultiplexes incoming data. In the other, demultiplexing is performed within the concentrator.

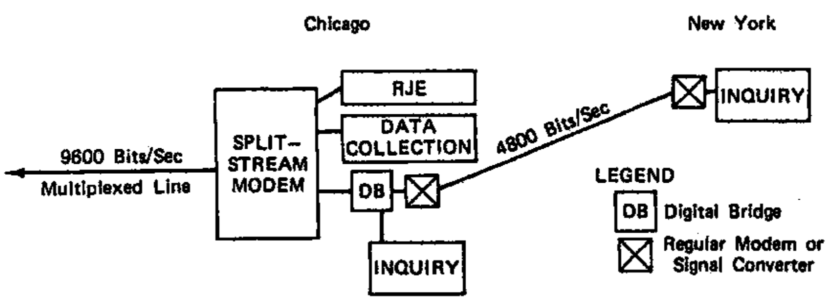


Figure 7.18. Solution for Case 3 employing digital bridge in Chicago.

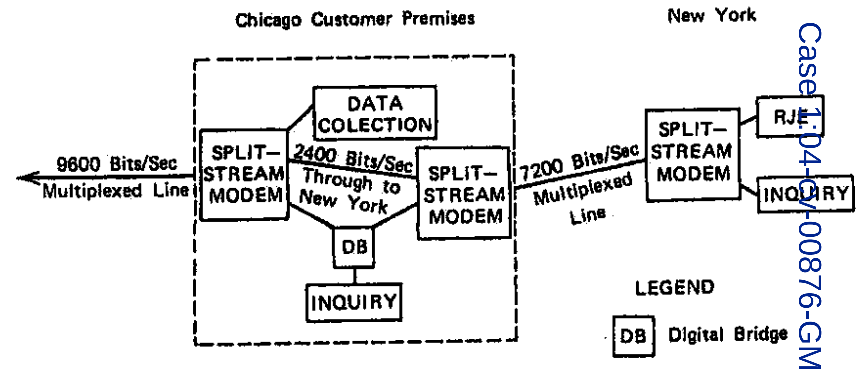


Figure 7.19. Solution for Case 4.

7.9. POSITIONING REMOTE MULTIPLEXERS AND CONCENTRATORS

It has previously been noted that the primary motivation for using multi-
plexing and concentration techniques is the reduction of total networ
costs. Obviously, the determination of the most suitable techniques an
locations of remote devices to accomplish the sharing constitutes a
important systems design problem that is closely related to the subject o

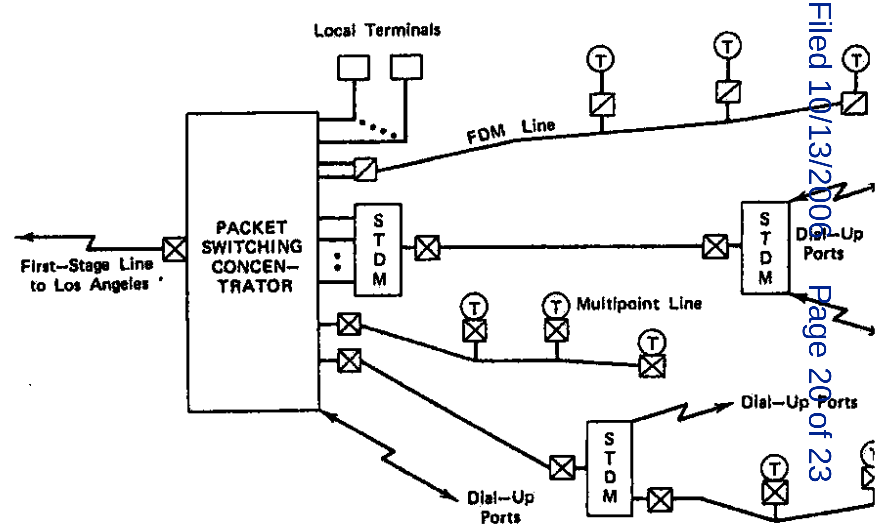


Figure 7.20. Example concentrator configuration illustrating termination of different types of second-stage lines.

network optimization, wherein topology, channel capacities, traffic, and performance criteria are jointly considered [15–17]. The details of network optimization being beyond the scope of this discussion, we conclude with a brief discussion of how such procedures may be applied to the multiplexer–concentrator site-location problem. Other network optimization problems are discussed in a paper by Frank and Chou [29].

Different approaches to the site-location problem have recently been proposed by Bahl and Tang [18], McGregor and Shen [28], and Doll et al. [19]. The Bahl and Tang paper describes a heuristic approach in which remote terminals are initially connected to many remote concentrators. All possible candidate sites initially contain concentrators. The algorithm removes a link between a concentrator and a remote terminal at each step until finally no more can be removed. It allows concentrators to die a graceful death instead of a violent one, as in alternative procedures. McGregor and Shen apply conventional operations research ideas for site-location positioning to the concentrator positioning problem. Doll et al. describe a heuristic interactive procedure that includes the topology between terminals and concentrators as a design variable. The essence of this procedure and the results of its application to a specific example are now summarized.

It is assumed that all remote terminal sites, central processing sites, and candidate sites for the multiplexers or concentrators to be positioned are given as inputs. The design procedure is based on the premise that remote line sharing devices will not be used unless they produce net cost savings in comparison to the best network without any multiplexers or concen-

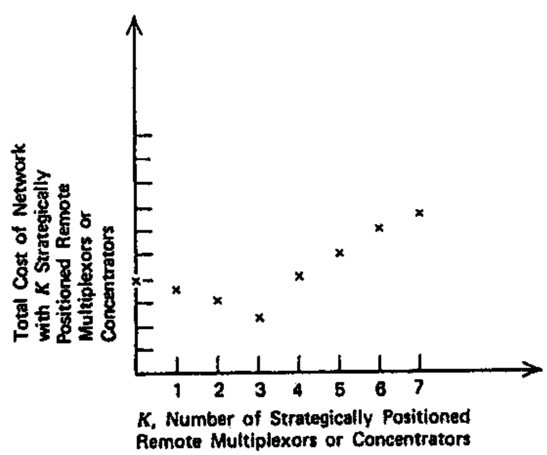


Figure 7.21. Typical cost curve of network with multiplexers of concentrators.

trators. Using an exhaustive search procedure, it effectively "finds" the best location for a first multiplexer. Then, assuming that a multiplexer exists at this site, it determines at which of the remaining candidate sites the second multiplexer should be positioned. The procedure is continued for as many iterations as desired, each subsequent iteration picking the best remaining site, on the assumption that any previously selected location will not be reconsidered. This restriction is currently imposed in the interest of computational feasibility and in order that topology can be considered as a variable. Research on other, possibly less restrictive variations of this theme is continuing.

Practical design experience to date has failed to produce any situations where manual improvements to the heuristically obtained solutions were possible, although no claim to true optimality is being made. Similarly, the use of this approach suggests that the cost of a network using multiplexers or concentrators tends to be a J-shaped function of the number of devices used, as shown in Figure 7.21. (In some networks, of course, it may cost more to use one multiplexer or concentrator than none.) The desired number of devices is given by the value of K in Figure 7.21 for which the total cost of the network is minimized. The geographic locations of these devices are also directly determined at consecutive steps in the iterations of the site selection routine.

Consider a nationwide network with terminal locations as shown in Figure 7.22 and a single computer center located in Chicago. It is assumed that the computer center in Chicago is to be fed by an unknown number (≤ 5) of remote STDMs and/or regionally positioned terminals with leased

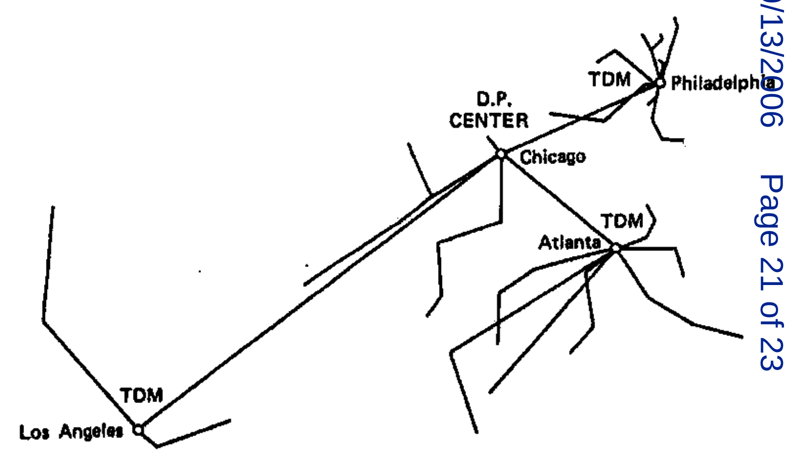


Figure 7.22. Example net.

Table 7.1. Results of Design Example for Varying Numbers of Remote STDMs

Number of STDMs K	Best Sites for Multiplexers	Monthly Cost of Network ($)
0		39,100
1	Los Angeles	35,200
2	Los Angeles, Philadelphia	32,400
3	Los Angeles, Philadelphia, Atlanta	31,700

point-to-point or multipoint lines.[a] We consider five possible sites for the placement of these remote STDMs—Atlanta, Los Angeles, Denver, New York, and Philadelphia.

Table 7.1 shows that Los Angeles is found to be the best site for the first STDM. Then, if Los Angeles is assumed as a fixed location for a multiplexer, Philadelphia is found to be the best of the remaining four sites for the second STDM. Finally, a third multiplexer is positioned at Atlanta, yielding total net savings of about 18%. The complete network configuration containing three STDMs is illustrated in Figure 7.22. Proportionately greater savings could reasonably be expected in applications involving relatively more terminals at substantial distances from the CPU site.

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[a]In the example, all links, excepting those from STDMs to the CPU, were costed using typical tariff rates for 150 bits/sec service. The multiplexed links were assumed to be conditioned voice-grade lines driven by commercially available modems. Commercially available STDMs having monthly rentals of $500 plus $30 for each low speed line termination were (arbitrarily) assumed in the cost calculations.

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EIGHT

LINE CONTROL PROCEDURES, NETWORK PROTOCOLS, AND CONTROL SOFTWARE